WHAT IS CLAIMED IS:

1	1. A hearing aid, comprising:
2	an input signal channel providing digital input signals;
3	a signal path adapted to process said digital input signals in accordance
4	with a predetermined signal processing algorithm to produce a digital output signal,
5	wherein said signal path further comprises at least one signal processing function
6	operating on a warped frequency scale; and
7	an output conversion means adapted to convert said output signals to an
8	audio output.
1	2. The hearing aid of claim 1, wherein said at least one signal
2	processing function further comprises a plurality of cascaded all-pass filters.
1	3. The hearing aid of claim 1, wherein said warped frequency scale
2	approximates a Bark scale.
1	4. A dynamic range compressor, comprising:
2	an input signal channel providing digital input signals;
3	a plurality of cascaded all-pass filters, wherein said digital input signals
4	pass through said plurality of cascaded all-pass filters, and wherein said plurality of
5	cascaded all-pass filters output a sequence of delayed samples;
6	means for applying a frequency domain transform on said sequence of
7	delayed samples, wherein a warped sequence results from said frequency domain
8	transform applying means;
9	means for calculating a plurality of frequency domain level estimates from
10	said warped sequence;
11	means for calculating a plurality of frequency domain gain coefficients
12	from said plurality of frequency domain level estimates;
13	means for applying an inverse frequency domain transform on said
14	plurality of frequency domain gain coefficients, wherein a set of compression filter
15	coefficients of a compression gain filter result from said inverse frequency domain
16	transform applying means; and

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means for convolving said sequence of delayed samples with said set of 17 18 compression filter coefficients to produce a digital output signal. 1 5. The dynamic range compressor of claim 4, further comprising a 2 hearing aid, wherein the dynamic range compressor is incorporated within said hearing 3 aid. 1 6. The dynamic range compressor of claim 4, wherein said plurality 2 of frequency domain gain coefficients comprise a warped time-domain filter. 1 7. The dynamic range compressor of claim 4, further comprising means for windowing said sequence of delayed samples, wherein a windowed sequence 2 3 of delayed samples results from said windowing means, and wherein said warped 4 sequence results from applying said frequency domain transform to said windowed 5 sequence of delayed samples. 1 8. The dynamic range compressor of claim 4, further comprising a digital-to-analog converter, said digital-to-analog converter converting said digital output 2 signals to analog output signals. 3 1 9. The dynamic range compressor of claim 8, further comprising an output transducer, said output transducer converting said analog output signals to an 2 3 audio output. 1 10. The dynamic range compressor of claim 4, said plurality of 2 cascaded all-pass filters comprising a plurality of first order all-pass filters. 1 11. The dynamic range compressor of claim 4, said sequence of delayed samples comprising 16 samples. 2

domain transform applying means, and said means for convolving said sequence of delayed samples.

digital processor, wherein said digital processor is adapted to provide said frequency domain transform applying means, said frequency domain level estimates calculating

means, said frequency domain gain coefficients calculating means, said inverse frequency

The dynamic range compressor of claim 4, further comprising a

I	13. The dynamic range compressor of claim 12, wherein said digital
2	processor comprises a software programmable digital signal processor.
1	14. The dynamic range compressor of claim 4, wherein said frequency
2	domain transform applying means uses a transform selected from the group consisting of
3	discrete Fourier transforms, fast Fourier transforms, Goertzel transforms, and discrete
4	cosine transforms.
1	15. The dynamic range compressor of claim 4, further comprising:
2	an input transducer, said input transducer converting audio input signals to
3	analog input signals; and
4	an analog-to-digital converter, said analog-to-digital converter converting
5	said analog input signals to said digital input signals.
1	16. The dynamic range compressor of claim 4, further comprising:
2	a digital-to-analog converter, said digital-to-analog converter converting
3	said digital output signals to analog output signals; and
4	an output transducer, said output transducer converting said analog output
5	signals to an audio output.
1	17. A dynamic range compressor, comprising:
2	an input signal channel providing digital input signals;
3	an input data buffer, said input data buffer holding at least one block of
4	data comprised of a portion of said digital input signals;
5	a plurality of cascaded all-pass filters, wherein a first block of said digital
6	input signals pass from said input data buffer through said plurality of cascaded all-pass
7	filters, and wherein said plurality of cascaded all-pass filters output a first sequence of
8	delayed samples;
9	means for windowing a first portion of said first sequence of delayed
10	samples, wherein a first windowed sequence of delayed samples results from said
11	windowing means;
12	means for applying a first frequency domain transform on said first
13	windowed sequence of delayed samples, wherein a first warped sequence results from
14	said first frequency domain transform applying means:

15	means for calculating a first plurality of frequency domain level estimates
16	of said first warped sequence;
17	means for windowing a second portion of said first sequence of delayed
18	samples, wherein a second windowed sequence of delayed samples results from said
19	windowing means;
20	means for applying a second frequency domain transform on said second
21	windowed sequence of delayed samples, wherein a second warped sequence results from
22	said second frequency domain transform applying means;
23	means for calculating a second plurality of frequency domain level
24	estimates of said second warped sequence;
25	means for summing said first and second plurality of frequency domain
26	level estimates, wherein a summed first and second plurality of frequency domain level
27	estimates results from said summing means;
28	means for normalizing said summed first and second plurality of frequency
29	domain level estimates, wherein a normalized first and second plurality of frequency
30	domain level estimates results from said normalizing means;
31	means for calculating a plurality of frequency domain gain coefficients
32	from said normalized first and second plurality of frequency domain level estimates;
33	means for applying an inverse frequency domain transform on said
34	plurality of frequency domain gain coefficients, wherein a set of compression filter
35	coefficients of a compression gain filter result from said inverse frequency domain
36	transform applying means;
37	means for convolving a second sequence of delayed samples with said
38	compression filter coefficients, said second sequence of delayed samples produced by a
39	second block of said digital input signals passing from said input data buffer through said
40	plurality of cascaded all-pass filters, wherein a digital output signal results from said
41	convolving means.
1	18. The dynamic range compressor of claim 17, further comprising a
2	hearing aid, wherein the dynamic range compressor is incorporated within said hearing
3	aid.
1	19. The dynamic range compressor of claim 17, wherein said plurality
2	of frequency domain gain coefficients comprise a warped time-domain filter.

1	20. The dynamic range compressor of claim 17, further comprising a
2	digital-to-analog converter, said digital-to-analog converter converting said digital output
3	signals to analog output signals.
1	21. The dynamic range compressor of claim 20, further comprising an
2	output transducer, said output transducer converting said analog output signals to an
3	audio output.
1	22. The dynamic range compressor of claim 17, said plurality of
2	cascaded all-pass filters comprising a plurality of first order all-pass filters.
1	23. The dynamic range compressor of claim 17, further comprising a
2	digital processor, wherein said digital processor is adapted to provide said windowing
3	means, said means for applying said first and second frequency domain transforms, said
4	means for calculating said first and second plurality of frequency domain level estimates,
5	said summing means, said normalizing means, said frequency domain gain coefficients
6	calculating means, said inverse frequency domain transform applying means, and said
7	convolving means.
1	24. The dynamic range compressor of claim 17, wherein said means
2	for applying said first and second frequency domain transforms use a transform selected
3	from the group consisting of discrete Fourier transforms, fast Fourier transforms, Goertzel
4	transforms, and discrete cosine transforms.
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1	25. The dynamic range compressor of claim 17, further comprising:
2	an input transducer, said input transducer converting audio input signals to
3	analog input signals; and
4	an analog-to-digital converter, said analog-to-digital converter converting
5	said analog input signals to said digital input signals.
1	26. The dynamic range compressor of claim 17 further comprising:
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3	a digital-to-analog converter, said digital-to-analog converter converting
	said digital output signals to analog output signals; and
4	an output transducer, said output transducer converting said analog output

signals to an audio output.

1	27. The hearing aid of claim 17, wherein said windowing means
2	provides a 50 percent overlap of said first and second pluralities of frequency domain
3	level estimates.
1	28. The hearing aid of claim 17, wherein a quantity of samples
2	corresponding to said first block of said digital input signals is equivalent to a quantity of
3	first order all-pass filters corresponding to said plurality of cascaded all-pass filters.
1	29. The hearing aid of claim 28, wherein said first portion of said first
2	sequence of delayed samples is comprised of a first half of said first sequence of delayed
3	samples and said second portion of said first sequence of delayed samples is comprised of
4	a second half of said first sequence of delayed samples.
1	30. A hearing aid, comprising:
2	an input signal channel providing digital input signals;
3	an input data buffer, said input data buffer holding a block of data of size
4	M comprised of a portion of said digital input signals;
5	a plurality of cascaded all-pass filters comprised of 2M cascaded all-pass
6	filters, wherein a first block of said digital input signals pass from said input data buffer
7	through said plurality of cascaded all-pass filters to form a first sequence of delayed
8	samples and wherein a second block of said digital input signals pass from said input data
9	buffer through said plurality of cascaded all-pass filters to form a second sequence of
10	delayed samples, and wherein said first sequence of delayed samples and said second
11	sequence of delayed samples form a combined sequence of delayed samples;
12	means for windowing a first portion of said combined sequence of delayed
13	samples, wherein said first portion is of size M, wherein a windowed sequence of delayed
14	samples results from said windowing means;
15	means for applying a 2M-point frequency domain transform on said
16	windowed sequence of delayed samples, wherein a warped sequence results from said
17	frequency domain transform applying means;
18	means for calculating a plurality of frequency domain level estimates of
19	said warped sequence;
20	means for calculating a plurality of frequency domain gain coefficients
21	from said plurality of frequency domain level estimates;

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means for applying an inverse frequency domain transform on said		
plurality of frequency domain gain coefficients, wherein a set of compression filter		
coefficients of a compression gain filter result from said inverse frequency domain		
transform applying means; and		
means for convolving a second portion of said combined sequence of		
delayed samples with said compression filter coefficients, wherein said second portion		

- layed samples with said compression filter coefficients, wherein said second portion is of size M, wherein a digital output signal results from said convolving means.
- 31. The dynamic range compressor of claim 30, further comprising a 2 hearing aid, wherein the dynamic range compressor is incorporated within said hearing aid.
 - 32. The dynamic range compressor of claim 30, wherein said plurality of frequency domain gain coefficients comprise a warped time-domain filter.
 - 33. The dynamic range compressor of claim 30, further comprising a digital-to-analog converter, said digital-to-analog converter converting said digital output signals to analog output signals.
 - 34. The dynamic range compressor of claim 33, further comprising an output transducer, said output transducer converting said analog output signals to an audio output.
 - 35. The dynamic range compressor of claim 30, said plurality of cascaded all-pass filters comprising a plurality of first order all-pass filters.
 - 36. The dynamic range compressor of claim 30, further comprising a digital processor, wherein said digital processor is adapted to provide said windowing means, said means for applying said 2M-point frequency domain transform, said means for calculating said plurality of frequency domain level estimates, said frequency domain gain coefficients calculating means, said inverse frequency domain transform applying means, and said convolving means.
 - The dynamic range compressor of claim 30, wherein said means 37. for applying said frequency domain transform uses a transform selected from the group

3	consisting of discrete Fourier transforms, fast Fourier transforms, Goertzel transforms,
4	and discrete cosine transforms.
1	38. The dynamic range compressor of claim 30, further comprising:
2	an input transducer, said input transducer converting audio input signals to
3	analog input signals; and
4	an analog-to-digital converter, said analog-to-digital converter converting
5	said analog input signals to said digital input signals.
1	39. The dynamic range compressor of claim 30, further comprising:
2	a digital-to-analog converter, said digital-to-analog converter converting
3	said digital output signals to analog output signals; and
4	an output transducer, said output transducer converting said analog output
5	signals to an audio output.
1	40. A method of processing sound in a hearing aid, comprising the
2	steps of:
3	receiving digital input signals;
4	passing a portion of said digital input signals through a plurality of
5	cascaded all-pass filters to form a sequence of delayed samples;
6	windowing said sequence of delayed samples;
7	applying a frequency domain transform to said windowed sequence of
8	delayed samples to form a warped sequence;
9	calculating a plurality of frequency domain level estimates from said
10	warped sequence;
11	calculating a plurality of frequency domain gain coefficients from said
12	plurality of frequency domain level estimates to form a warped time domain filter;
13	applying an inverse frequency domain transform on said plurality of
14	frequency domain gain coefficients to form a set of compression filter coefficients; and
15	convolving said sequence of delayed samples with said compression filter
16	coefficients to form a digital output signal.